ARO 24373.20-MA-SDI

DAAL03 - 86 - K - 0108
20 March 1990

Adaptive, Robust, High - Resolution Signal Processing

Final Report

BRINGING KNOWLEDGE TO BEAR

Donald W. Tufts
Principal Investigator
The University of Rhode Island
Department of Electrical Engineering

This Research was sponsored by the SDIO/IST, managed by the Army Research Office

18. (continued)

Robust Detection, Adaptive Detection

19. (continued)

Research results obtained during the three and one-half year period ending in January 1990 are summarized. And the papers are referenced in which the detailed results appear. The summary is separated into the following three areas:

1. The Arithmetic Fourier Transform (AFT)

The Arithmetic Fourier Transform (AFT) is a new algorithm for Fourier analysis, narrowband filtering and beamforming. From the point of view of Applied Mathematics, especially the areas of Approximation Theory and Numerical Analysis, the AFT is a set of quadrature formulas by which the integrals which define Fourier Coefficients can be exactly evaluated for band-limited functions. This approach is based on the number-theoretic method of Mobius inversion.

The discovery of the AFT was motivated by the goal of efficient, high-speed calculation of Fourier coefficients in a special-purpose integrated circuit. The computations of the AFT can be carried out in parallel, pipelined channels, and the individual operations are very simple to execute and to control. Except for one scaling for each channel, all the operations are additions or subtractions. Also, analog switched-capacitor realizations of the AFT have been studied.

The AFT has also been modified to compute the Discrete Cosine Transform (DCT) and extended to calculate two-dimensional Fourier coefficients. Thus, the AFT can be applied to the representation and processing of images and to multidimensional array processing.

2. Analysis of Performance of High-Resolution Signal Processing

We have developed methods and formulas for calculating the statistics of high-resolution estimates of parameters of multiple, simultaneous signals in gaussian noise, at both high and low values of signal-to-noise ratio (SNR). These results provide insights, and understanding in addition to the numerical results. We have derived the first results for threshold SNR, that is the value of SNR below which the statistics of the estimation errors become much worse than those predicted by the Cramer-Rao bounds.

Also, we have provided a simpler and more widely applicable analysis of the performance of our previously proposed Principal Component Inverse (PCI) method of interference suppression in adaptive detection. More specifically, we derive an approximate formula for the probability density of SNR at the output of the adaptive filter/beamformer.

3. Extending the Applicability of High-Resolution Signal Processing.

We have shown how high-resolution methods can be applied to signal detection in the presence of impulsive components; how our previously presented minimum-norm algorithm can be applied to sensor arrays with arbitrary geometry; and how high-resolution techniques can be applied to adaptive equalization for high-speed data communication. In addition, we developed a new method for more detailed analysis of voiced sounds and for more robust signal analysis.

Adaptive, Robust, High-Resolution Signal Processing
Final Report
DAAL-03-86-K-0108
Donald W. Tufts, Principal Investigator

Research results obtained during the three and one-half year period ending in January 1990 are summarized. And the papers are referenced in which the detailed results appear. The summary is separated into the following three areas:

1. The Arithmetic Fourier Transform (AFT)

The Arithmetic Fourier Transform (AFT) [26,12,18,17]is a new algorithm for Fourier analysis, narrowband filtering and beamforming. From the point of view of Applied Mathematics, especially the areas of Approximation Theory and Numerical Analysis, the AFT is a set of quadrature formulas by which the integrals which define Fourier Coefficients can be exactly evaluated for band-limited functions.[1] This approach is based on the number-theoretic method of Mobius inversion.

The discovery of the AFT was motivated by the goal of efficient, high-speed calculation of Fourier coefficients in a special-purpose integrated circuit .[9] The computations of the AFT can be carried out in parallel, pipelined channels, and the individual operations are very simple to execute and to control. Except for one scaling for each channel, all the operations are additions or subtractions. Also, analog switched-capacitor realizations of the AFT have been studied. [8,25]

The AFT has also been modified to compute the Discrete Cosine Transform (DCT) and extended to calculate two-dimensional Fourier coefficients. [24] Thus, the AFT can be applied to the representation and processing of images and to multidimensional array processing.

2. Analysis of Performance of High-Resolution Signal Processing

We have developed methods and formulas for calculating the statistics of high-resolution estimates of parameters of multiple, simultaneous signals in gaussian noise, at both high and low values of signal-to-noise ratio (SNR). [2,4,5,11,14,15] These results provide insights, and understanding in addition to the numerical results. We have derived the first results for threshold SNR, [20,27],that is the value of SNR below which the statistics of the estimation errors become much worse than those predicted by the Cramer-Rao bounds.

Also, we have provided a simpler and more widely applicable analysis of the performance of our previously proposed Principal Component Inverse (PCI) method of interference suppression in adaptive detection. More specifically, we derive an approximate formula for the probability density of SNR at the output of the adaptive filter/beamformer. [6,21]

3. Extending the Applicability of High-Resolution Signal Processing.

We have shown how high-resolution methods can be applied to signal detection in the presence of impulsive components; how our previously presented minimum-norm algorithm can be applied to sensor arrays with arbitrary geometry; [3] and how high-resolution techniques can be applied to adaptive equalization for high-speed data communication. [7,10,16,23] In addition, we developed a new method for more detailed analysis of voiced sounds [19] and for more robust signal analysis. [21,22].

REFERENCES

1) G.F. Boudreaux-Bartels, D.W. Tufts, P. Dhir, G. Sadasiv, and G. Fischer, "Analysis of Errors in the Computation of Fourier Coefficients Using the Arithmetic Fourier Transform (AFT) and Summation by Parts (SBP)", Proceedings of IEEE ICASSP '89 pp 1011-14, Glasgow, Scotland, May 23-26, 1989.

2) D.W. Tufts, R.J. Vaccaro and A.C. Kot, "Analysis of Estimation of Signal Parameters by Linear Prediction at High SNR Using Matrix Approximation," Proceedings of IEEE ICASSP '89 pp 2194-98, Glasgow, Scotland, May 23-26, 1989.

3) Fu Li, R.J. Vaccaro, and D.W. Tufts, "Min-Norm Linear Prediction for Arbitrary Sensor Arrays", Proceedings of IEEE ICASSP '89 pp 2613-16, Glasgow, Scotland, May 23-26, 1989.

4) R.J. Vaccaro and Fu Li, "MUSIC Performance Prediction By Matrix Approximation at High SNR", Proceedings of 1989 Conference on Information Sciences and Systems, pp 477-482, John Hopkins University, Baltimore, MD, March 1989.

5) Fu Li, R.J. Vaccaro, and D.W. Tufts, "Unified Performance Analysis of Subspace-Based Estimation Algorithms," Proceedings of IEEE ICASSP '90, pp , Albuquerque, NM April 3-6, 1990.

6) I. Kirsteins and D.W. Tufts, "Rapidly Adaptive Nulling of Interference", to appear as a chapter in the book "Lecture Notes in Control and Information Sciences," Springer/Verlag/ Publishers.

7) M. Barton and D.W. Tufts, "Reduced-Rank Least Squares Channel Estimation," accepted for publication IEEE Trans. on ASSP.

8) G. Fischer, D.W. Tufts, and R. Unnikrishnan, "VLSI Implementation of the Arithmetic Fourier Transform," Proceedings of IEEE Midwest Symposium on Circuits and Systems. Aug. '89.

9) M.T. Shih, I.S. Reed, T.K. Truong, E. Hendon and D.W. Tufts, "A VLSI Architecture for Simplified Fourier Transform Algorithm," Submitted for publication to IEEE Trans. on ASSP.

10) M. Barton and D. W. Tufts, "A Suboptimum Linear Receiver Derived from SVD Based Channel Estimates," Proceedings of ICC '89 pp 264.1-264.5.

11) A.C. Kot, D. W. Tufts, and R.J. Vaccaro, "Analysis of Linear Prediction by Matrix Approximation," Submitted for publication to IEEE Transactions.

12) I.S. Reed, D.W. Tufts, T.K. Truong, M.T. Shih X. Yin and X. Yu, "Fourier Analysis and Signal Processing by Use of the Mobius Inversion Formula," IEEE Trans. on ASSP, Mar. 1990, pp 458-469

14) Fu Li and R.J. Vaccaro, "Analytical Performance Comparison of the Min-Norm and MUSIC Algorithms with Arbitrary Array Geometry," Accepted for publication in IEEE Trans. on Aerospace and Electronic Systems.

15) Fu Li, R.J. Vaccaro and D.W. Tufts, "Performance Analysis of the State Space Realization (TAM) and ESPRIT Algorithms for DOA Estimation, "Submitted for publication to IEEE Trans. on Antennas and Propagation.

16) M. Barton and D. W. Tufts, "A Suboptimum Linear Receiver Based on a Parametric Channel Model," accepted for publications as a Correspondence Item, IEEE Transactions on Communications.

17) D.W. Tufts, Comments on "A Note on the Computational Complexity of the Arithmetic Fourier Transform," IEEE, ASSP, July 1989, pp 1147.

18) D.W. Tufts and G. Sadasiv, "Arithmetic Fourier Transform & Adaptive Delta Modulation: A Symbiosis for High-Speed Computation," Proceedings of the SPIE, Innovative Science & Technology Symposium, Vol. 880, pp 168-178, January 11-12, 1988, Los Angeles, CA.

19) S. Parthasarathy and D.W. Tufts, "Signal Modeling by Exponential Segments and Application in Voiced Speech Analysis," Proceedings of IEEE ICASSP '87, Vol 2, pp 645-648, April 6-9, 1987.

20) D.W. Tufts, A. Kot and R.J. Vaccaro, "The Threshold Analysis of SVD-Based Algorithms," Proceedings of IEEE ICASSP '88, pp 2416-2419, April 11-14, 1988, New York City, NY.

21) C.D. Melissinos, D. W. Tufts and I. P. Kirsteins, "Very Fast Adaptive Array Processing," Proceedings of the 25th IEEE Conf. on Decision and Control, Athens, Greece, Dec. 1986.

22) D.W. Tufts and C.D. Melissinos, "Threshold Extension of SVD-Based Algorithms," Proceedings of IEEE ICASSP '88, pp 2825-2828, April 11-14, 1988, New York City, NY.

23) D.W. Tufts, M. Barton and A. Efron, "Robust, Adaptive Filtering for Data Transmission," Proceedings of IEEE ICASSP '87, Vol 4, pp 2121-2124, April 6-9, 1987.

24) D.W. Tufts, Z. Fan and Z. Cao, "Image Processing and the Arithmetic Fourier Transform, Proceedings of SPIE/IST Conference on High Speed Computing II '89, Vol 1058 pp 46-53, January 17-18, 1989, Los Angeles, CA.

25) D.W. Tufts, and G. Fischer, "VLSI Implementation of the Arithmetic Fourier Transform (AFT): A New Approach to High-Sped Computation for Signal Processing," Chapter 25 pp 264-275 of VLSI Signal Processing, III, edited by R.W. Brodersen and H. S. Moscovitz, IEEE Press, New York

26) D.W. Tufts, and G. Sadasiv, "The Arithmetic Fourier Transform," IEEE ASSP Magazine, January 1988 pp 13-17.

27) D.W. Tufts, A. Kot and R.J. Vaccaro, "The Threshold Analysis of SVD-Based Algorithms," Proceedings of Twenty-first Annual Asilomar Conference on Signals, Systems and Computers, Nov. 1987.